CLAIMS

This patent reveals the beamforming method and system at reception with a new technique called Progressive Focal Correction technique and its Variable option. According to said method and system, the coherent composition of signals coming from sources located along a selected direction and arriving to a set of N transducers receptors or elements spatially distributed, that form an array. A beam former at reception performs the coherent composition of the received signals that are coming from discrete sources located along said direction. The coherent composition of the signals originated at each point and received by each of the N elements of the array provide one sole output signal that represents the signal that would be obtained with a sole receptor transducer, of the same size of the array, individually focused on each and every one of the foci, a technique known in the art as dynamic focusing with deflection.

The signals received are amplified and analogically conditioned before being digitalized. The progressive focal correction technique, with or without the variable option, and the system that performs it as described in this patent are applied to the digitalized signals by means of an analogic-digital converter. The set of devices that process the signal received by an element of the array until it is combined with those corresponding to other elements is called a channel.

Contrary to other methods that are part of the prior art, the method and system that are the object of this patent are not based on delaying the signals received by each of the array elements, nor are interpolation processes used. The system and method are based in the sampling of the received signals at the instants or arrival to each of the elements of the array from each of the foci. The method guaranties that the composition of samples is coherent, that is, only those that correspond to the same focus are combined, and also non-redundant, since only those that are going to be used in the composition process are acquired. The system requires a non-uniform signal sampling clock that must also be different for each of the channels.

On the other hand, and contrary to other techniques in which each sampling clock is established by a '1' and '0' sequence in an individual memory for each channel, or is calculated in real time by means of specific digital circuits based on the values of a diversity of parameters that have been programmed during acquisition, the method and system that are the object of the present invention, the sampling instants are established by interpreting the content of tables associated to each one of the channels, that are calculated an stored in memories before initiation the signal acquisition process, and making the loading of multiple parameters during the acquisition time unnecessary. Also, the amount of storage required is limited because the focal correction to be applied to the sampling clock to obtain the next focus is codified. Moreover, each focal correction code can be shared by multiple consecutive samples. Finally, the architecture of the technique that is patented herein, presents a high regularity, which facilitates its integration in VLSI devices and its modular embodiment.

The following statement-claims are claimed:

1. A method for beam forming at reception called Progressive Focal Technique intended for applications in which a diversity of N transducers grouped in an array receive vibratory energy from foci F_i, $i=0,\ 1,\ 2\ldots$, located along a programmable direction, wherein it samples the signal received in each channel at the arrival instant from each focus with an absolute error lower than half a period of a master clock with a T_x period, in such a manner that the consecutive samples e_{ki} $i=0,\ 1,\ 2,\ldots$.obtained in each channel k correspond to the consecutive values of the signal originated at each focus F_i, i=0, 1, 2,..., ensuring that each sample of the resulting signal r_i, i=0, 1, 2,... is obtained by the sum of the N samples e_{ki} k=1, 2,..., N:

$$r_i = \sum_{k=1}^{N} e_{ki} \qquad i = 0, 1, 2, \ldots$$

or of a linear combination such as:

$$r_i = \sum_{k=1}^{N} A_k e_{ki} \qquad i = 0, 1, 2, \ldots$$

where the Ak coefficients are fixed or programmable values, operation which is called summation or coherent composition of the N signals.

2. A method of beam forming at reception according to Claim 1 wherein the receptor channel that corresponds to the k element of the array, that includes circuits for the conditioning and amplifying of the signal, the sampling instant that corresponds to the first focus F_0 is determined in non real time as based on the number of cycle $N_A(k)$ of the master clock that are between the origin of times selected with the constraint that all the times be positive and the arrival instant of the F_0 focus to channel k, in such a manner that if the flight time of the signal that corresponds to the F_i focus to the k element is $T_i\ (k)$, calculated as based in the geometry of the system and the speed of propagation of the waves in the medium, the $N_A(k)$ value is given by:

$$N_A(k) = \left[\frac{T_0(k)}{T_X} \right]_{\uparrow\downarrow}$$

Where I*I↑↓ represents the function of rounding to the nearest whole number that determines $N_A(k)$ with an absolute error lower than $T_x/2$, a function that can be done by means of a $N_A(k)$ master clock cycle counter, which generates an acquisition signal pulse in channel k at the time in which said

counter overloads; just as it has been describe in the present patent application, there are various manners to achieve the physical embodiment, the preferred one being an embodiment in which the counter is unfolded in two parts, one of small size that is actuated by the master clock and the main part that is actuated by a clock which frequency is a submultiple of that of the master clock.

3. A method for beam forming at reception according to any of the previous Claims wherein for other foci F_i $i > 0$, the sampling instant at channel k is determined in real time based on the Q_{ki} value of b bits, stored in a focal correction memory associated to said channel k that codifies the advance to be applied to the nominal value v of master clock periods that is between two consecutive foci, where the Q_{ki} code is calculated in non real time based on the difference ΔT_{ki} between the arrival instant to the k element of the signals coming from the focus i-1 and the focus i as:

$$Q_{ki} = \text{v} - \left[\frac{\Delta T_{ki}}{T_X}\right]_{\uparrow\downarrow}$$

Where I*I↑↓ represents the function of rounding to the nearest whole number and v is the nominal interval between foci expressed in master clock periods, the Q_{ki} codes corresponding to successive foci F_h i=1, 2, 3,..., being programmed consecutively in a focal correction memory associated to the k channel which contents are read in signal acquisition time to produce the variable advance at the sampling instant for each focus F_i as a function of the Q_{ki} value by means of digital circuits of which several examples have been given in the present description report based on the shift registers of programmable length, other alternatives being possible, such as the use of programmable counters, without it modifying in any manner the foundational basic idea described in this Claim.

4. A method and system for beam forming at reception according to any of the previous Claims wherein it allows inserting a fixed or programmable number

m of samples between foci by means of a programmable counter module $N_M=m$ that is incremented every time that a new sample is acquired, and when it returns to zero the focal correction memory access address is incremented, making possible that a total of m samples share the same focal correction code Q_{ki} , which makes possible increasing the efficiency of the encoding process that may be a fraction of a bit per acquired sample while the sampling instant error is maintained basically bounded within half a period of the master clock since it has digital circuits that distribute uniformly the Q_{ki} advance amongst the samples m that are acquired between foci, circuits that are based on a table in which, before acquisition and for each possible value of $Q_{ki} \leq 2^b-1$, a total of m bits g are programmed, and indicate for each of the m samples if the acquisition instant of said sample must be advanced $(g=1)$ or not $(g=0)$ by 1 master clock cycle, it being a particular case of the methodology exposed in the previous Claims.

5. A method and system for beam forming at reception according to any of the previous Claims wherein the distance between foci can be varied by means of the acquisition of a variable number of samples m between foci that is incremented from an initial value m_0 when the value of a focal shift code J_i, of 1 bit associated to the F_i focus is J_i =1, an option called Variable Progressive Focal Correction technique, that increases the efficiency of the use of the focal correction memory by increasing the distance between foci with time, where the J_i value is common to all the channels for each value of i, that can be tabulated in a completely centralized manner for all the channels, in a partially centralized manner affecting a set of channels, as it is preferably done, or individually per channel, which allows establishing a balance between flexibility of application and memory efficiency; this can be implemented by simply substituting the register of the constant value m associated to each k channel by a counter that is loaded with the initial value m_0 before initiating each acquisition and, during the acquisition time, said counter is incremented each time that signals are received from a new focus F_i and the $J_i=1$ code is received.

6. A method and system for beam forming at reception according to any of the previous Claims wherein it has counters to determine the number of foci N_F

and of samples N_s to be acquired, considering that each focus may include the obtention of m samples, m being constant or variable so when the N_F counter is overloaded the progressive focal correction mechanisms and the focal shift update mechanism are blocked, freezing also, the memory address that serves to access the focal correction codes in each channel, although the signal acquisition process may continue until the counter N_s that governs the number of samples to be acquired reaches the terminal value, facilitating that one sole memory per channel can store the focal correction codes for multiple angles of deflection, multiple configurations of the active aperture or a combination of both, the initial focal correction memory address can be programmed, at any rate, before each acquisition or be automatically obtained if the acquisition sequence corresponds to the order in which the focal correction codes have been stored for the current application.

7. A method and system for beam forming at reception according to any of the previous Claims wherein it can operate optionally with a dynamic aperture, activating a counter associated to each channel *k* in which a number of output samples *Nz(k)* of null value is programmed from the beginning of the acquisition process to be combined with the corresponding samples of the remaining channels, and therefore for these *Nz(k)* samples the result is equivalent to eliminating the *k* element in the composition of the corresponding output samples, which effectively performs the dynamic aperture system.

8. A method and system for beam forming at reception according to any of the previous Claims wherein it can insert an apodization function or multiplication of the values of each acquired sample times by constant that can be programmed for each channel.

9. A method and system for beam forming at reception according to any of the previous Claims wherein it performs the coherent composition of the signal by means of a combination of distributed queues or FIFO memories and adders, preferably forming a tree structure in which each two FIFOs are associated to a 2 entry adder.

10. A method and system for beam forming at reception according to any of the previous Claims wherein it facilitates its modular construction with a flexible architecture that allows for the coherent composition of an indefinite number of signals from the perspective of the logic, and for which certain number of channels are grouped to form a sub-module, several sub-modules configure a module and a system is configured by several modules, where all the channels are similar and process the acquired information in parallel.

11. A method and system for beam forming at reception according to any of the previous Claims wherein it uses the methodology and constructive models described in this specification and shown in the attached figures below.